AMENDMENT TO THE CLAIMS:

1. (Previously Presented) A method for determining whether to accept a new call to be routed from a first gateway to a second gateway in an IP network, the method comprising the steps of:

obtaining, at the first gateway, information indicative of the quality of service of voice calls being transmitted from the first gateway to the second gateway via a plurality of network paths between the first gateway and the second gateway;

determining, using at least a portion of said information, a plurality of congestion status parameters indicative of respective congestion statuses of the network paths, each of said network paths being associated with respective first gateway egress interfaces and a second gateway system IP address; and

determining, using at least one of the congestion status parameters, whether to accept the new call into the network at the first gateway for transmission toward the second gateway via one of the network paths.

2. (Previously Presented) The method of claim 23, wherein the new call is accepted into the IP network at a reduced bandwidth in the case of the congestion status parameter associated with the one of the network paths exceeding a lower threshold.

3. (Previously Presented) The method of claim 23, wherein the new call is not accepted into the IP network in the case of the congestion status parameter associated with the one of the network paths exceeding the upper threshold.

4. (Previously Presented) The method of claim 1, wherein the obtained information comprises, for each of at least one of the network paths, a number of sent packets transmitted from the first gateway to the second gateway via the network path, wherein the number of sent packets comprises a number of lost packets, a number of late packets and a number of received packets.

5. (Previously Presented) The method of claim 1, wherein the obtained information comprises, for each of at least one of the network paths, a delay of received packets transmitted from the first gateway to the second gateway via the network path.

6. (Previously Presented) The method of claim 1, wherein the obtained information comprises, for each of at least one of the network paths, a delay variation of received packets transmitted from the first gateway to the second gateway via the network path.

7-8. (cancelled)

9. (Previously Presented) The method of claim 1, wherein, for at least one of the network paths, the congestion status parameter of the network path is identified as a packet lost ratio (PLR).

10. (Previously Presented) The method of claim 9, wherein PLR is defined as

$$PLR = \frac{(\text{lost packets} + \text{late packets})}{(\text{received packets} + \text{lost packets} + \text{late packets})}.$$

11. (Previously Presented) The method of claim 2, wherein bandwidth is reduced for the new call by selecting a first encoder to encode the new voice call information in a bandwidth that is smaller than bandwidths of other calls accepted in the network that are encoded by a second encoder.

12. (Previously Presented) The method of claim 2, wherein the bandwidth of the new call is reduced by increasing the packet size for the new call, wherein the packet size is indicative of a size of a corresponding voice sample.

13. (Previously Presented) The method of claim 2, wherein the bandwidth of the new call is reduced by activating the characteristic of silence suppression for said newly accepted voice call.

14. (Previously Presented) Apparatus comprising a first gateway for interfacing voice call data from a public switch telephone network to an Internet Protocol (IP) network, said first gateway comprising:

a first circuit for passing the voice call data of voice calls to the internet protocol network;

a second circuit for receiving quality-of-service information associated with voice calls currently being transmitted toward a second gateway via the first circuit; and

a third circuit for:

calculating, based on the received quality-of-service information, a plurality of congestion status parameters associated with the respective network paths between the first gateway and the second gateway, wherein the congestion status parameters are indicative of respective congestion statuses of the network paths, each of said network paths being associated with respective first gateway egress interfaces and a second gateway system IP address; and

determining, using at least one of the congestion status parameters, whether a new voice call is to be accepted into the IP network via the first circuit for transmission toward the second gateway via one of the network paths.

15. (Previously Presented) The apparatus of claim 14, wherein said first circuit further comprises one or more Ethernet cards that are connected to the internet protocol network.

16. (Previously Presented) The apparatus of claim 14, wherein said second circuit is at least one strongarm card.

17. (Previously Presented) The apparatus of claim 16, wherein the strongarm card is connected to at least one Ethernet card via a host CPU circuit.

18. (Previously Presented) The apparatus of claim 14, wherein the third circuit determines whether the new voice call is to be accepted into the internet protocol

network via the first circuit by comparing each of the at least one of the congestion status parameters to at least one threshold.

19. (Previously Presented) The apparatus of claim 14, wherein, for at least one of the network paths, the congestion status parameter is a packet loss ratio defined as

$$\mathrm{PLR} = \frac{(\text{lost packets} + \text{late packets})}{(\text{received packets} + \text{lost packets} + \text{late packets})}.$$

20. (Previously Presented) The apparatus of claim 19, wherein, for at least one of the network paths, the third circuit compares the packet loss ratio to a lower threshold and if the packet loss ratio is less than the lower threshold, the new voice call is accepted into the internet protocol network.

21. (Previously Presented) The apparatus of claim 19, wherein, for at least one of the network paths, the third circuit compares the packet loss ratio to the lower threshold and an upper threshold, and if lower threshold < packet loss ratio < upper threshold, the new voice call is accepted into the internet protocol network at a reduced bandwidth.

22. (Previously Presented) The apparatus of claim 19, wherein, for at least one of the network paths, the third circuit compares the packet loss ratio to the upper threshold, and if the packet loss ratio is greater than the upper threshold, the new voice call is blocked from entering the internet protocol network.

23. (Previously Presented) The method of claim 1, further comprising:

accepting the new call into the IP network at the first gateway for transmission toward the second gateway via one of the network paths, wherein the new call is accepted into the IP network in the case of the congestion status parameter associated with the one of the network paths not exceeding an upper threshold.

24. (Previously Presented) The method of claim 1, wherein the information indicative of the quality of service of voice calls being transmitted from the first gateway to the

second gateway comprises a plurality of performance reports associated with the voice calls, wherein determining the congestion status parameters of the network paths comprises:

determining, for each of the performance reports, one of the network paths with which the performance report is associated; and

determining, for each of the network paths, the congestion status parameter of the network path using at least a portion of the performance reports determined to be associated with the network path.

25. (Previously Presented) The method of claim 1, wherein the information indicative of the quality of service of voice calls being transmitted from the first gateway to the second gateway comprises a plurality of performance reports associated with the voice calls, wherein determining the congestion status parameters of the network paths comprises:

for each of at least one of the network paths:

selecting only a subset of the performance reports associated with the network path; and

determining the congestion status parameter of the network path using the selected subset of the performance reports associated with the network path.

26. (previously presented) The method of claim 1, wherein determining whether to accept the new call into the network at the first gateway is made using call control logic, wherein the call control logic is updated using the congestion status parameters.

27. (Previously Presented) The method of claim 26, wherein, for each of at least one of the network paths, the call control logic is updated using the congestion status parameter for the network path periodically.

28. (Previously Presented) The method of claim 26, wherein, for each of at least one of the network paths, the call control logic is updated using the congestion status parameter for the network path on an exception reporting basis.

29. (Previously Presented) The method of claim 1, wherein the first gateway comprise a plurality of ports associated with the respective plurality of network paths, wherein determining the congestion status parameters comprises:

for each network path:

for each of at least one voice call being transmitted from the first gateway to the second gateway via the network path, computing a congestion status value associated with the voice call using the obtained information associated with the voice call; and

determining the congestion status parameter of the network path using the at least one congestion status value computed for the network path.

30. (Previously Presented) The method of claim 29, wherein, for each of at least one of the network paths, the congestion status parameter for the network path is determined by selecting the congestion status value computed for the network path that is indicative of the greatest amount of congestion on the network path.

31. (Previously Presented) The method of claim 1, wherein the determination as to whether to accept the new call into the network at the first gateway is performed using all of the network congestion parameters.

32. (Previously Presented) The method of claim 31, further comprising:

accepting the new call into the IP network at the first gateway for transmission toward the second gateway via one of the network paths, wherein the one of the network paths for the new call is the one of the network paths having the associated congestion status parameter indicative of the least amount of congestion.

33. (Previously Presented) The method of claim 1, wherein determining whether to accept the new call into the network at the first gateway comprises:

for each of at least one of the network paths, updating a call admission control policy for the network path based on the congestion status parameter determined for the network path; and

determining whether to accept the new call into the network at the first gateway based on the updated call admission control policy.